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| APPLICATION NO. | FILING DATE | FIRST NAMED INVENTOR | ATTORNEY DOCKET NO. | CONFIRMATION NO. |
|---|---|---|---|---|
| 10/519,000 | 12/21/2004 | Carl Christensen | PU020292 | 6971 |

24498 7590 08/12/2008

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| EXAMINER | |
|---|---|
| COLUCCI, MICHAEL C | |

| ART UNIT | PAPER NUMBER |
|---|---|
| 2626 | |

| MAIL DATE | DELIVERY MODE |
|---|---|
| 08/12/2008 | PAPER |

**Please find below and/or attached an Office communication concerning this application or proceeding.**

The time period for reply, if any, is set in the attached communication.

| Office Action Summary | Application No. | Applicant(s) | |
|---|---|---|---|
| | 10/519,000 | CHRISTENSEN ET AL. | |
| | Examiner | Art Unit | |
| | MICHAEL C. COLUCCI | 2626 | |

***-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --***

**Period for Reply**

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

**Status**

1)☐ Responsive to communication(s) filed on _____.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

**Disposition of Claims**

4)☒ Claim(s) *1-22* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-22* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

**Application Papers**

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on *14 January 2008* is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

**Priority under 35 U.S.C. § 119**

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

**Attachment(s)**

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☒ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application

6) ☐ Other: _____.

## DETAILED ACTION

### *Continued Examination Under 37 CFR 1.114*

1. A request for continued examination under 37 CFR 1.114, including the fee set forth in 37 CFR 1.17(e), was filed in this application after final rejection. Since this application is eligible for continued examination under 37 CFR 1.114, and the fee set forth in 37 CFR 1.17(e) has been timely paid, the finality of the previous Office action has been withdrawn pursuant to 37 CFR 1.114. Applicant's submission filed on 06/13/2008 has been entered.

### ***Response to Arguments***

2. Applicants arguments with respect to claims 1-22 have been considered but are moot in view of the new grounds of rejection. Examiner has withdrawn the reference of Lew and has incorporated a new reference: Pillay et al. US 20030195645 A1 (hereinafter Pillay). Pillay teaches the extraction of data from a biphase encoded audio stream using a time window, wherein subframes and preambles are present within the audio stream. Pillay also teaches the estimation of bit sample length as well as extraction of audio data from an audio stream, wherein a time window is used. Pillay accomplishes the same limitations disclosed within claims 1-22, even with the use of PLL's to decode a biphase signal and extract frames.

### ***Claim Rejections - 35 USC § 103***

3. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

**4.** Claim 1-4, 8, 11-18, and 20-22 are rejected under 35 U.S.C. 103(a) as being unpatentable over Lew, US 5245667 A (hereinafter Lew) in view of Pillay et al. US 20030195645 A1 (hereinafter Pillay) and further in view of Gillick et al US 4837831 A (hereinafter Gillick).

Re claims 1,and 11, Lew teaches a method of extracting digital audio data words from a serialized stream of digital audio data (Col. 4 lines 34-68 & Fig. 2), comprising:

constructing a timing window from an estimated bit time for said serialized stream of digital audio data, said timing window having a preamble sub-window (Col. 6 lines 16-32) and at least one data sub-window (Col. 4 lines 34-68 & Fig. 2);

extracting plural digital audio data words from said serialized stream of digital audio (col. 1 line 22-28) based upon the location of each transition in said serialized stream of digital audio data relative to said preamble sub-window (Col. 6 lines 16-32) and said at least one data sub-window of said timing window (Col. 4 lines 34-68 & Fig. 2);

each one of said extracted plural digital audio data words having a preamble identifiable by a combination of at least one transition located in said preamble sub-window (Col. 6 lines 16-32) of said timing window and at least one transition located in said at least one data sub-window of said timing window.

However, Lew fails to teach a preamble identifiable by a combination of at least one transition located in said preamble sub-window of said timing window and at least one transition located in said at least one data sub-window of said timing window.

Pillay teaches a method of extracting a clock from a biphase encoded bit stream includes the step of detecting a stream of samples each having a sample size measured between consecutive bit phase transitions. A sample length is determined for each sample, the sample length approximating a number of least common multiples in the corresponding sample size. A preamble is detected from the sample lengths of a

sequence of the samples and decoded to determine an expected logic level of the clock following a transition at an expected clock edge. The expected level of the clock is gated with the biphase encoded data to generate a control signal in advance of the opening of the time window (Pillay Abstract).

Additionally, Pillay teaches a logic level transition of the data bit at each active edge of the bit clock, otherwise it is considered to be an error in the encoding scheme. (With the exception that preambles by definition include one or more of such biphase errors.) FIG. 7 illustrates a portion of a typical AES/EBU (SPD/IF) data stream. The stream is divided into blocks each composed of 192 frames (Frames 0-191). Each frame in turn is composed a pair of subframes, each including Channel A and Channel B data, along with one of three types of 4-bit preambles. An X preamble precedes each Channel A subframe (except at the beginning of the block), a Y preamble precedes each Channel B subframe and a Z preamble precedes each Channel A subframe at the beginning of the block (Pillay [0061]).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Lew to incorporate a preamble identifiable by a combination of at least one transition located in said preamble sub-window of said timing window and at least one transition located in said at least one data sub-window of said timing window as taught by Pillay to allow for the reduction of biphase errors when extracting and generating clock and control signals, wherein frames, subframes, and preambles are present within the time window (Pillay [0061]).

However, Lew in view of Pillay fails to teach extracted plural digital audio data words.

Gillick teaches that the acquisition of multiple utterances of each vocabulary word, method 100 advances to step 106. This step performs a plurality of substeps 108, 110, and 112 for each word in the vocabulary. The first of these substeps, step 108, itself comprises two substeps, 114 and 116, which are performed for each utterance of each word. Step 114 finds an anchor for each utterance, that is, the first location in the utterance at which it has attained a certain average threshold amplitude. Step 116 calculates five smoothed frames for each utterance, positioned relative to its anchor. Additionally, Gillick teaches in reference to figures 4 and 5, where FIG. 4 schematically represents how such smoothed frames are calculated. A smoothed frame 118 is calculated from five individual frames 104A-104E, of the type described above with regard to FIG. 3. According to this process, each pair of successive individual frames 104 are averaged, to form one second level frame 120. Thus the individual frames 104A and 104B are averaged to form the second level frame 120A, and the individual frames 104B and 104C are averaged to form the second level frame 120B, and so on, as is shown in FIG. 4 (Gillick Col. 8 lines 5-37).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Lew in view of Pillay to incorporate extracted plural digital audio data words within an audio stream as taught by Gillick to allow for the detection and location of multiple utterances within an audio signal, wherein the repetition of an utterance is not limited to adjacent words and can have a

separation between the repeated words, where transitions from a repeated word to the next can be smoothed and the location of occurrence can be stored in memory as to detect a repeating segment of multiple words (i.e. chorus) (Gillick Col. 8 lines 5-37).

Re claims 2 and 15, Lew teaches a pair of successive transitions (Col. 8 lines 24-42) located in said preamble sub-window followed by a pair of successive transitions located in said at least one data sub-window (Col. 4 lines 34-68 & Fig. 2).

However, Lew fails to teach a preamble sub-window followed by a pair of successive transitions

Pillay teaches a method of extracting a clock from a biphase encoded bit stream includes the step of detecting a stream of samples each having a sample size measured between consecutive bit phase transitions. A sample length is determined for each sample, the sample length approximating a number of least common multiples in the corresponding sample size. A preamble is detected from the sample lengths of a sequence of the samples and decoded to determine an expected logic level of the clock following a transition at an expected clock edge. The expected level of the clock is gated with the biphase encoded data to generate a control signal in advance of the opening of the time window (Pillay Abstract).

Additionally, Pillay teaches a logic level transition of the data bit at each active edge of the bit clock, otherwise it is considered to be an error in the encoding scheme. (With the exception that preambles by definition include one or more of such biphase

errors.) FIG. 7 illustrates a portion of a typical AES/EBU (SPD/IF) data stream. The stream is divided into blocks each composed of 192 frames (Frames 0-191). Each frame in turn is composed a pair of subframes, each including Channel A and Channel B data, along with one of three types of 4-bit preambles. An X preamble precedes each Channel A subframe (except at the beginning of the block), a Y preamble precedes each Channel B subframe and a Z preamble precedes each Channel A subframe at the beginning of the block (Pillay [0061]).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Lew to incorporate preamble sub-window followed by a pair of successive transitions located in said at least one data sub-window as taught by Pillay to allow for the reduction of biphase errors when extracting and generating clock and control signals, wherein frames, subframes, and preambles are present within the time window (Pillay [0061]).

However, Lew in view of Pillay fails to teach the method of claim 1, and further comprising identifying said extracted data words as having a first type of preamble if said extracted data words have a pair of successive transitions.

Gillick teaches that the acquisition of multiple utterances of each vocabulary word, method 100 advances to step 106. This step performs a plurality of substeps 108, 110, and 112 for each word in the vocabulary. The first of these substeps, step 108, itself comprises two substeps, 114 and 116, which are performed for each utterance of each word. Step 114 finds an anchor for each utterance, that is, the first

location in the utterance at which it has attained a certain average threshold amplitude. Step 116 calculates five smoothed frames for each utterance, positioned relative to its anchor. Additionally, Gillick teaches in reference to figures 4 and 5, where FIG. 4 schematically represents how such smoothed frames are calculated. A smoothed frame 118 is calculated from five individual frames 104A-104E, of the type described above with regard to FIG. 3. According to this process, each pair of successive individual frames 104 are averaged, to form one second level frame 120. Thus the individual frames 104A and 104B are averaged to form the second level frame 120A, and the individual frames 104B and 104C are averaged to form the second level frame 120B, and so on, as is shown in FIG. 4 (Gillick Col. 8 lines 5-37).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Lew in view of Pillay to incorporate extracted plural digital audio data words within an audio stream as taught by Gillick to allow for the detection and location of multiple utterances within an audio signal, whererin the repetition of an utterance is not limited to adjacent words and can have a separation between the repeated words, where transitions from a repeated word to the next can be smoothed and the location of occurrence can be stored in memory as to detect a repeating segment of multiple words (i.e. chorus) (Gillick Col. 8 lines 5-37).

Re claims 3 and 16, Lew teaches preamble sub-window (Col. 6 lines 16-32) separated by a pair of successive transitions located in said at least one data sub-window .

However, Lew fails to teach the method of claim 2, and further comprising identifying said extracted data words as having a second type of preamble if said extracted data words have a pair of non-successive transitions.

Pillay teaches a method of extracting a clock from a biphase encoded bit stream includes the step of detecting a stream of samples each having a sample size measured between consecutive bit phase transitions. A sample length is determined for each sample, the sample length approximating a number of least common multiples in the corresponding sample size. A preamble is detected from the sample lengths of a sequence of the samples and decoded to determine an expected logic level of the clock following a transition at an expected clock edge. The expected level of the clock is gated with the biphase encoded data to generate a control signal in advance of the opening of the time window (Pillay Abstract).

Additionally, Pillay teaches a logic level transition of the data bit at each active edge of the bit clock, otherwise it is considered to be an error in the encoding scheme. (With the exception that preambles by definition include one or more of such biphase errors.) FIG. 7 illustrates a portion of a typical AES/EBU (SPD/IF) data stream. The stream is divided into blocks each composed of 192 frames (Frames 0-191). Each frame in turn is composed a pair of subframes, each including Channel A and Channel B data, along with one of three types of 4-bit preambles. An X preamble precedes each Channel A subframe (except at the beginning of the block), a Y preamble precedes each Channel B subframe and a Z preamble precedes each Channel A subframe at the beginning of the block (Pillay [0061]).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Lew to incorporate identifying said extracted data words as having a second type of preamble if said extracted data words have a pair of non-successive transitions as taught by Pillay to allow for the reduction of biphase errors when extracting and generating clock and control signals, wherein frames, subframes, and preambles are present within the time window (Pillay [0061]).

Gillick teaches that the acquisition of multiple utterances of each vocabulary word, method 100 advances to step 106. This step performs a plurality of substeps 108, 110, and 112 for each word in the vocabulary. The first of these substeps, step 108, itself comprises two substeps, 114 and 116, which are performed for each utterance of each word. Step 114 finds an anchor for each utterance, that is, the first location in the utterance at which it has attained a certain average threshold amplitude. Step 116 calculates five smoothed frames for each utterance, positioned relative to its anchor. Additionally, Gillick teaches in reference to figures 4 and 5, where FIG. 4 schematically represents how such smoothed frames are calculated. A smoothed frame 118 is calculated from five individual frames 104A-104E, of the type described above with regard to FIG. 3. According to this process, each pair of successive individual frames 104 are averaged, to form one second level frame 120. Thus the individual frames 104A and 104B are averaged to form the second level frame 120A, and the individual frames 104B and 104C are averaged to form the second level frame 120B, and so on, as is shown in FIG. 4 (Gillick Col. 8 lines 5-37).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Lew in view of Pillay to incorporate extracted plural digital audio data words within an audio stream as taught by Gillick to allow for the detection and location of multiple utterances within an audio signal, whererin the repetition of an utterance is not limited to adjacent words and can have a separation between the repeated words, where transitions from a repeated word to the next can be smoothed and the location of occurrence can be stored in memory as to detect a repeating segment of multiple words (i.e. chorus) (Gillick Col. 8 lines 5-37).

Re claims 4 and 17, Lew teaches the method of claim 3, and further comprising identifying said extracted data words as having a third type of preamble (Col. 6 lines 16-32) if said extracted data words have a transition located in said preamble sub-window followed by first, second and third transitions located in said at least one data sub-window.

However, Lew fails to teach a third type of preamble if said extracted data words have a transition located in said preamble sub-window followed by first, second and third transitions located in said at least one data sub-window

Pillay teaches a method of extracting a clock from a biphase encoded bit stream includes the step of detecting a stream of samples each having a sample size measured between consecutive bit phase transitions. A sample length is determined for each sample, the sample length approximating a number of least common multiples in

the corresponding sample size. A preamble is detected from the sample lengths of a sequence of the samples and decoded to determine an expected logic level of the clock following a transition at an expected clock edge. The expected level of the clock is gated with the biphase encoded data to generate a control signal in advance of the opening of the time window (Pillay Abstract).

Additionally, Pillay teaches a logic level transition of the data bit at each active edge of the bit clock, otherwise it is considered to be an error in the encoding scheme. (With the exception that preambles by definition include one or more of such biphase errors.) FIG. 7 illustrates a portion of a typical AES/EBU (SPD/IF) data stream. The stream is divided into blocks each composed of 192 frames (Frames 0-191). Each frame in turn is composed a pair of subframes, each including Channel A and Channel B data, along with one of three types of 4-bit preambles. An X preamble precedes each Channel A subframe (except at the beginning of the block), a Y preamble precedes each Channel B subframe and a Z preamble precedes each Channel A subframe at the beginning of the block (Pillay [0061]).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Lew to incorporate a third type of preamble if said extracted data words have a transition located in said preamble sub-window followed by first, second and third transitions located in said at least one data sub-window as taught by Pillay to allow for the reduction of biphase errors when extracting and generating clock and control signals, wherein frames, subframes, and preambles are present within the time window (Pillay [0061]).

Re claims 8 and 18, Lew teaches the method of claim 1, wherein said estimated bit time is derived from said serialized stream of digital audio data (Col. 4 lines 34-68 & Fig. 2).

Re claims 12 and 13, Lew teaches the method of claim 11, wherein said fast sample rate is at least about twenty times faster than a data rate for said serialized stream of digital audio data (Lew Col. 5 line 44-53).

Re claims 14 and 21, Lew teaches the method of claim 13, wherein each one of said extracted plural digital audio data words has a preamble (Col. 6 lines 16-32) identifiable by a combination of at least one transition located in said preamble sub-window (Col. 6 lines 16-32) of said timing window-and at least one transition located in said at least one data sub-window of said timing window (Col. 4 lines 34-68 & Fig. 2).

However, Lew fails to teach at least one transition located in said preamble sub-window of said timing window-and at least one transition located in said at least one data sub-window of said timing window.

Pillay teaches a method of extracting a clock from a biphase encoded bit stream includes the step of detecting a stream of samples each having a sample size measured between consecutive bit phase transitions. A sample length is determined for each sample, the sample length approximating a number of least common multiples in the corresponding sample size. A preamble is detected from the sample lengths of a

sequence of the samples and decoded to determine an expected logic level of the clock following a transition at an expected clock edge. The expected level of the clock is gated with the biphase encoded data to generate a control signal in advance of the opening of the time window (Pillay Abstract).

Additionally, Pillay teaches a logic level transition of the data bit at each active edge of the bit clock, otherwise it is considered to be an error in the encoding scheme. (With the exception that preambles by definition include one or more of such biphase errors.) FIG. 7 illustrates a portion of a typical AES/EBU (SPD/IF) data stream. The stream is divided into blocks each composed of 192 frames (Frames 0-191). Each frame in turn is composed a pair of subframes, each including Channel A and Channel B data, along with one of three types of 4-bit preambles. An X preamble precedes each Channel A subframe (except at the beginning of the block), a Y preamble precedes each Channel B subframe and a Z preamble precedes each Channel A subframe at the beginning of the block (Pillay [0061]).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Lew to incorporate at least one transition located in said preamble sub-window of said timing window-and at least one transition located in said at least one data sub-window of said timing window as taught by Pillay to allow for the reduction of biphase errors when extracting and generating clock and control signals, wherein frames, subframes, and preambles are present within the time window (Pillay [0061]).

Re claim 20, Lew teaches a bi-phase decoder for use in decoding a stream of AES-3 digital audio data, comprising:

a decoder circuit coupled to receive a stream of AES-3 digital audio data, an estimated bit time for said stream of AES-3 digital audio data (Col. 4 lines 34-68 & Fig. 2) and a fast clock, said fast clock having a frequency of about at least twenty times faster than a frequency of said stream of AES-3 digital audio data (Lew Col. 5 line 44-53);

a data store (Col. 8 line 24-42) coupled to said decoder circuit, said data store receiving sub frames of digital audio data extracted, from said stream of AES-3 digital (Col. 4 lines 34-68 & Fig. 2) audio data by said decoder circuit (Fig. 1 & Col. 3 line 55-65);

said decoder circuit extracting sub frames of said digital audio data by constructing a timing window from said estimated bit time (Col. 4 lines 34-68 & Fig. 2), sampling said stream of AES-3 digital audio data using said fast clock (Lew Col. 5 line 44-53) and applying said sampled stream of AES-3 digital audio data to said timing window to identify transitions (Col. 6 lines 16-32), in said sampled stream of AES-3 digital audio data, indicative of preambles of said sub frames of digital audio data.

However, Lew fails to teach extracting sub frames of said digital audio data by constructing a timing window

sampled stream of AES-3 digital audio data, indicative of preambles of said sub frames of digital audio data

Pillay teaches a method of extracting a clock from a biphase encoded bit stream includes the step of detecting a stream of samples each having a sample size measured between consecutive bit phase transitions. A sample length is determined for each sample, the sample length approximating a number of least common multiples in the corresponding sample size. A preamble is detected from the sample lengths of a sequence of the samples and decoded to determine an expected logic level of the clock following a transition at an expected clock edge. The expected level of the clock is gated with the biphase encoded data to generate a control signal in advance of the opening of the time window (Pillay Abstract).

Additionally, Pillay teaches a logic level transition of the data bit at each active edge of the bit clock, otherwise it is considered to be an error in the encoding scheme. (With the exception that preambles by definition include one or more of such biphase errors.) FIG. 7 illustrates a portion of a typical AES/EBU (SPD/IF) data stream. The stream is divided into blocks each composed of 192 frames (Frames 0-191). Each frame in turn is composed a pair of subframes, each including Channel A and Channel B data, along with one of three types of 4-bit preambles. An X preamble precedes each Channel A subframe (except at the beginning of the block), a Y preamble precedes each Channel B subframe and a Z preamble precedes each Channel A subframe at the beginning of the block (Pillay [0061]).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Lew to incorporate extracting sub frames of said digital audio data by constructing a timing window and sampling a stream of AES-3

digital audio data, indicative of preambles of said sub frames of digital audio data as taught by Pillay to allow for the reduction of biphase errors when extracting and generating clock and control signals, wherein frames, subframes, and preambles are present within the time window (Pillay [0061]).

Re claim 22, Lew teaches the apparatus of claim 21, and further comprising a bit time estimator circuit having an input coupled to receive said stream of AES-3 digital (Col. 4 lines 34-68 & Fig. 2) audio data and an output coupled to said decoder circuit (Col. 4 lines 34-68 & Fig. 3), said bit time estimator determining said estimated bit time for output to said decoder circuit (Col. 4 lines 34-68 & Fig. 3).

5. **Claims 5-7 are rejected under 35 U.S.C. 103(a) as being unpatentable over Lew US 5245667 A (hereinafter Lew) in view of Pillay et al. US 20030195645 A1 (hereinafter Pillay) and Gillick et al US 4837831 A (hereinafter Gillick) and further in view of Akagiri US 5490130 (hereinafter Akagiri).**

Re claims 5-7, Lew teaches the method of claim 4, wherein said timing window is construed such that said at least one data sub-window includes a first data sub-window (Col. 4 lines 34-68 & Fig. 2).

However, Lew in view of Pillay and Gillick fails to teach a sub-window which extends from about ¼ times said estimated bit time to about ¾ times said estimated bit time and a second data sub window which extends from about ¾ times said estimated bit time to about 1¼ times said estimated bit time.

NOTE: The use of *about* is construed to be an estimate with no fixed range or deviation limitation, where 1.5 or even 2 can be considered close to .5 without a specific variation constraint and is therefore construed to functionally equivalent to a scaling of .25 or .5.

Akagiri teaches that frequency range signals is then divided in time into blocks to which block floating processing and orthogonal transform processing is applied. The block length decision circuit 45 adaptively determines the block length of the blocks in each of the frequency ranges according to dynamic characteristics of the digital input signal. The digital input signal is notionally divided in time into frames. Then, after the digital input signal is divided into plural frequency range signals, each frequency range signal is divided into the blocks in which the frequency range signal will be orthogonally transformed. Each block corresponds to a frame or an integral fraction (e.g., 1/2, 1/4) of a frame. Thus, the maximum block length in which each frequency range signal is orthogonally transformed is equal to the frame length (Akagiri Col. 15 lines 7-20).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Lew in view of Pillay and Gillick to incorporate a window or frame that is scaled by about .25 to 1.25 as taught by REFB to allow for orthogonal constraints to be met during the transformation of a signal from the time to frequency range as to not overlap data between adjacent frames by extending/shortening a frame (Akagiri Col. 15 lines 7-20).

**6. Claims 9-10 and 19 are rejected under 35 U.S.C. 103(a) as being unpatentable over Lew US 5245667 A (hereinafter Lew) in view of Pillay et al. US 20030195645 A1 (hereinafter Pillay) and Gillick et al US 4837831 A (herein after Gillick) and further in view of Tackin US 7180892 (hereinafter Tackin).**

Re claims 9-10 and 19, Lew teaches the method of claim 18, and further comprising:

identifying transitions in said serialized stream of digital audio data which occur within said constructed bit window (Col. 4 lines 34-68 & Fig. 2),

However, Lew fails to teach the time separating a set of successive identified transitions being a measurement of said estimated bit time.

Gillick teaches that the acquisition of multiple utterances of each vocabulary word, method 100 advances to step 106. This step performs a plurality of substeps 108, 110, and 112 for each word in the vocabulary. The first of these substeps, step 108, itself comprises two substeps, 114 and 116, which are performed for each utterance of each word. Step 114 finds an anchor for each utterance, that is, the first location in the utterance at which it has attained a certain average threshold amplitude. Step 116 calculates five smoothed frames for each utterance, positioned relative to its anchor. Additionally, Gillick teaches in reference to figures 4 and 5, where FIG. 4 schematically represents how such smoothed frames are calculated. A smoothed frame 118 is calculated from five individual frames 104A-104E, of the type described above with regard to FIG. 3. According to this process, each pair of successive individual frames 104 are averaged, to form one second level frame 120. Thus the

individual frames 104A and 104B are averaged to form the second level frame 120A, and the individual frames 104B and 104C are averaged to form the second level frame 120B, and so on, as is shown in FIG. 4 (Gillick Col. 8 lines 5-37).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Lew in view of Pillay to incorporate time separating a set of successive identified transitions being a measurement of said estimated bit time as taught by Gillick to allow for the detection and location of multiple utterances within an audio signal, whererin the repetition of an utterance is not limited to adjacent words and can have a separation between the repeated words, where transitions from a repeated word to the next can be smoothed and the location of occurrence can be stored in memory as to detect a repeating segment of multiple words (i.e. chorus) (Gillick Col. 8 lines 5-37).

However, Lew in view of Pillay and Gillick fails to teach estimating minimum and maximum bit window times; constructing a bit window from said minimum and maximum bit window

determining said estimated bit time from a running average of plural measurements of said estimated bit time (Tackin Col. 26 line 53 - Col. 27 line 9).

Tackin teaches voice synchronizer should operate with or without sequence numbers, time stamps, and SID packets. The voice synchronizer should also operate with voice packets arriving out of order and lost voice packets. In addition, the voice synchronizer preferably provides a variety of configuration parameters which can be

specified by the host for optimum performance, including minimum and maximum target holding time. With these two parameters, it is possible to use a fully adaptive jitter buffer by setting the minimum target holding time to zero msec and the maximum target holding time to 500 msec (or the limit imposed due to memory constraints). Although the preferred voice synchronizer is fully adaptive and able to adapt to varying network conditions, those skilled in the art will appreciate that the voice synchronizer can also be maintained at a fixed holding time by setting the minimum and maximum holding times to be equal. These estimates are periodically quantized and transmitted in a SID packet by the comfort noise estimator (usually at the end of a talk spurt and periodically during the ensuing silent segment, or when the background noise parameters change appreciably). The comfort noise estimator 81 should update the long running averages, when necessary, decide when to transmit a SID packet, and quantize and pass the quantized parameters to the packetization engine 78 (Tackin Col. 36 lines 15-31).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Lew in view of Pillay and Gillick to incorporate estimating minimum and maximum bit window times; constructing a bit window from said minimum and maximum bit window and determining said estimated bit time from a running average of plural measurements of said estimated bit time as taught by Tackin to allow for a maximum and minimum time to produce a buffer having a reduced amount of jitter when extracting and quantizing information from a signal, wherein the use a running/moving average for time based data can be smoothed,

reducing the number of fluctuations based on a maximum and minimum period (Tackin Col. 36 lines 15-31).

### *Conclusion*

7. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure. US 6405093 B1, US 6628999 B1, US 6782300 B2.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Michael C. Colucci whose telephone number is (571)-270-1847. The examiner can normally be reached on 9:30 am - 6:00 pm, Monday-Friday.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Richemond Dorvil can be reached on (571)-272-7602. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

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